FORM PTO-1390 (Modified) (REV 11-98) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTORNEY'S DOCKET NUMBER 112740-218
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371	U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/868398

JC18 Rec'd PCT/PTO 18 JUN 2001

INTERNATIONAL APPLICATION NO.	INTERNATIONAL FILING DATE	PRIORITY DATE CLAIMED
PCT/DE99/03838	**01 December 1999**	**17 December 1998**

TITLE OF INVENTION

METHOD AND SYSTEM FOR CHANNEL CODING AND DECODING OF INFORMATION STRUCTURED IN FRAMES

APPLICANT(S) FOR DO/EO/US

Joachim Hagenauer et al.

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))
   a. ☒ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☒ A copy of the International Search Report (PCT/ISA/210).
8. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))
   a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☒ have been transmitted by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☐ have not been made and will not be made.
9. ☒ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
10. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).
11. ☒ A copy of the International Preliminary Examination Report (PCT/IPEA/409).
12. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).

Items 13 to 20 below concern document(s) or information included:

13. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
14. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
15. ☒ A **FIRST** preliminary amendment.
16. ☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.
17. ☒ A substitute specification.
18. ☐ A change of power of attorney and/or address letter.
19. ☒ Certificate of Mailing by Express Mail
20. ☒ Other items or information:

Submission of Drawings - Figures 1-6 on six sheets

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/868598	INTERNATIONAL APPLICATION NO. PCT/DE99/03838	ATTORNEY'S DOCKET NUMBER 112740-218

21. The following fees are submitted:.

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

JC18 Rec'd PCT/PTO 18 JUN 2001

	CALCULATIONS	PTO USE ONLY
☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2) paid to USPTO and International Search Report not prepared by the EPO or JPO $1,000.00		
☒ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but Internation Search Report prepared by the EPO or JPO $860.00		
☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO $710.00		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) but all claims did not satisfy provisions of PCT Article 33(1)-(4) $690.00		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(1)-(4) $100.00		
ENTER APPROPRIATE BASIC FEE AMOUNT =	$860.00	
Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)).	$0.00	

CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE		
Total claims	9 - 20 =	0	x $18.00	$0.00	
Independent claims	2 - 3 =	0	x $80.00	$0.00	
Multiple Dependent Claims **(check if applicable).**			☐	$0.00	
	TOTAL OF ABOVE CALCULATIONS		=	$860.00	
Reduction of 1/2 for filing by small entity, if applicable. Verified Small Entity Statement must also be filed (Note 37 CFR 1.9, 1.27, 1.28) **(check if applicable).**			☐	$0.00	
		SUBTOTAL	=	$860.00	
Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)).			+	$0.00	
		TOTAL NATIONAL FEE	=	$860.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable).**			☐	$0.00	
		TOTAL FEES ENCLOSED	=	$860.00	
				Amount to be: refunded	$
				charged	$

☒ A check in the amount of **$860.00** to cover the above fees is enclosed.

☐ Please charge my Deposit Account No. in the amount of to cover the above fees. A duplicate copy of this sheet is enclosed.

☒ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. **02-1818** A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

William E. Vaughan (Reg. No. 39,056))
Bell, Boyd & Lloyd LLC
P.O. Box 1135
Chicago, Illinois 60690

SIGNATURE

William E. Vaughan
NAME

39, 056
REGISTRATION NUMBER

June 18, 2001
DATE

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IN THE UNITED STATES ELECTED/DESIGNATED OFFICE
OF THE UNITED STATES PATENT AND TRADEMARK OFFICE
UNDER THE PATENT COOPERATION TREATY-CHAPTER II

PRELIMINARY AMENDMENT

APPLICANTS:	Joachim Hagenauer et al.	DOCKET NO: 112740-218
SERIAL NO:		GROUP ART UNIT:
		EXAMINER:
INTERNATIONAL APPLICATION NO:		PCT/DE99/03838
INTERNATIONAL FILING DATE:		01 December 1999
INVENTION:	METHOD AND SYSTEM FOR CHANNEL CODING AND DECODING OF INFORMATION STRUCTURED IN FRAMES	

Assistant Commissioner for Patents,
Washington, D.C. 20231

Sir:

Please amend the above-identified International Application before entry into the National stage before the U.S. Patent and Trademark Office under 35 U.S.C. §371 as follows:

In the Specification:

Please replace the Specification of the present application, including the Abstract, with the following Substitute Specification:

SPECIFICATION

TITLE

METHOD AND SYSTEM FOR CHANNEL CODING AND DECODING OF INFORMATION STRUCTURED IN FRAMES

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates, generally, to a method and a system for channel coding and decoding of information structured in frames and, more particularly, to such a method and system directed to adaptive multirate coding.

Description of the Prior Art

Source signals and source information such as voice, audio, picture and video signals virtually always contain statistical redundancy, that is to say redundant information. This redundancy can be greatly reduced by source coding, thus allowing efficient transmission and storage of the source signal. This reduction in redundancy gets rid of signal contents which were redundant before transmission and are based on prior knowledge of, for example, statistical parameters in the signal profile. The bit rate of the source-coded information (source bits or data bits) is also referred to as the source bit rate. During the source decoding after transmission, these components are added to the signal once again, so that virtually no loss of quality can be verified objectively.

On the other hand, it is normal for signal transmission to add redundancy deliberately by channel coding once again, in order largely to correct for the influence of channel interference on the transmission. Additional redundant bits thus make it possible for the receiver or decoder to identify errors, and possibly also to correct them. The bit rate of the channel-coded information is also referred to as the gross bit rate.

In order to allow information, in particular voice data, picture data or other user data, to be transmitted as efficiently as possible by the limited transmission capacities of a transmission medium, in particular a radio interface, this information to be transmitted is compressed by source coding before transmission, and is protected against channel errors by channel coding. Various methods are, in each case, known for doing this. For example, in the GSM (Global System for Mobile Communication) System, voice can be coded by a full rate voice codec a half rate voice codec, or an enhanced full rate voice codec.

For the purposes of this application, the terms voice codec or coding also refer to a method for encoding and/or for corresponding decoding. This method

may also cover sources and/or channel coding and can be applied to data other than voice data.

In the course of the further development of the European mobile radio standard GSM, a new Standard for coded voice transmission is being developed, which will allow the overall data rate and the splitting of the data rate between the source coding and channel coding to be set adaptively depending on the channel state and the network conditions (system load). Instead of the voice codecs described above, which have a fixed source bit rate, new voice codecs are intended to be used for this purpose, whose source bit rate is variable and is matched to changing frame conditions for information transmission. The main aims of such AMR (Adaptive Multirate) voice codecs are to achieve landline network quality for voice transmission in various channel conditions, and to ensure optimum distribution of the channel capacity, taking account of specific network parameters. After carrying out a conventional source coding method, the compressed information is in structured form, in frames, in which case the source bit rate may differ from frame to frame depending on the code mode being used.

In order to achieve standard gross bit rates, the information contained within a frame is channel-coded in a different manner depending on the source bit rate or code mode, particularly at a different rate, in such a manner that the gross bit rate after channel coding corresponds to the selected channel mode (half rate or full rate). For example, such an AMR voice codec can operate using the half rate (HR) channel in good channel conditions and/or in highly loaded radio cells. When the channel conditions are poor, a dynamic change should be made to the full rate (FR) channel, and vice versa. Within such a channel mode (half rate or full rate), various code modes are available for different voice and channel coding rates, and these are likewise selected to match the channel quality (rate adaptation). In the process, the gross bit rate after channel coding remains constant within one channel mode (22.8 kbps for the full rate channel FR and 11.4 kbps for the half-rate channel HR). This is intended to result in the best voice quality, taking account of the changing channel conditions. Thus, with such adaptive coding, different rates are used for

voice coding (variable source bit rate) depending on the channel conditions in a transmission path, on the requirements for specific network parameters, and depending on the voice. Since the gross bit rate after channel coding is intended to remain constant, an appropriately adapted variable number of error protection bits are added during channel coding.

In order to decode such variably coded information after transmission, it is helpful for information about the coding method used at the transmission end, particularly the source bit rate and/or the type of channel coding used at the transmission end, to be known at the receiving end. For this purpose, it is possible for certain bits, so-called mode bits, to be generated at the transmission end which, for example, indicate the rate used for source or channel coding.

It is known for the mode bits to be protected and to be transmitted, using a block code, independently of the source bits (data bits). In consequence, these so-called mode bits can be decoded first, with the source bits being determined subsequently, depending on this first decoding result. A disadvantage of this method is that the error frequency is relatively high in the mode bits since, particularly in mobile radio channels which are subject to fading, the correction capability in the decoder is low, owing to the short block length.

Alternatively, it is possible to carry out the decoding in a number of steps. To do this, decoding is initially carried out using a first mode, and a CRC (Cyclic Redundancy Check) is used to determine whether this mode was worthwhile. If this is not the case, decoding is carried out using a further mode, and the result is checked once again. This method is repeated with all the modes until a sensible result is obtained. The disadvantage of this method is the high computation complexity, which leads to both an increased power consumption and a decoding delay.

It is known from US 5537410 for information to be transmitted in a frame in order to indicate a rate.

The present invention is, thus, based on the problem of specifying a method and a system for channel coding and decoding which allows information about the type of coding to be transmitted in a simple manner and reliably.

SUMMARY OF THE INVENTION

According to the present invention, therefore, a method is specified in which a first portion of first information items, for example user information items, is channel coded in a standard manner independently of the nature of the coding, and for different types of coding.

This ensures that a first portion of the first information items also can be used for decoding second information items for describing the coding of first information items, and better quality error correction for the second information items can ~~thus~~ be carried out by the increase in the block length, associated therewith, of the convolution codes used for coding the second information items. This allows multiple decoding, using the try and error principle described above, to be avoided.

Information for describing the coding of first information items may, in this case, contain information for describing the source coding and/or the channel coding and/or other first information items for decoding, such as the type of coding (source and/or channel coding of the first information items) or the coding rate (source and/or channel coding of the first information items).

The present invention can be used advantageously, particularly if the coding of the information can be carried out such that it is adapted to different types.

In one embodiment of the present invention, the rate of channel coding of at least a second portion of the first information items, for example the user information, is matched to the quality of the transmission channel and/or to the network load. It is thus possible to match the channel coding to changing frame conditions in a communications system, and to transmit this adaptation at the transmission end to a receiver in a simple and reliable manner.

In further embodiments, the second information items contain signaling information and/or information for describing the reception quality, in order to

influence a transmitter as a function of the reception result at that time. It is thus possible to control the transmission of information based on the principle of a control loop.

For channel coding, it is advantageous to use convolution codes, and to match the length of the first portion of the first information items, which is channel-coded in a standard manner, at least approximately to the length of influence of the convolution code being used.

Furthermore, the problem is solved by a method for decoding of information structured in frames, in which a first portion of the first information items is also used for decoding second information items. This allows the coded transmission of second information items with a sufficiently long block length, and avoids complex multiple decoding based on the trial and error principle described above.

Decoding carried out in a manner such as this is advantageous, in particular, if one of the methods described above has been used to code the information at the transmission end as part of the information transmission process.

The problem is also solved by systems for channel coding and decoding of information structured in frames, in which a digital signal processor is in each case set up in such a manner that a first portion of the first information items can be channel-coded in a standard manner, independently of the nature of the coding, for different types of coding, and/or in such a manner that a first portion of the first information items can also be used for decoding second information items. These arrangements are particularly suitable for carrying out the methods according to the present invention, or one of the alternative embodiments explained above.

In this case, digital transmission of information is described, in particular. Nevertheless, the present invention can also be applied to the storage of information, since the writing of information to a storage medium and the reading of information from a storage medium corresponds, in terms of the present invention, to the transmission of information and the reception of information.

DESCRIPTION OF THE DRAWINGS

Figure 1 shows an outline circuit diagram of a mobile radio system;

Figure 2 shows a schematic illustration of major elements of a telecommunications transmission chain;

Figure 3 shows a schematic illustration of an adaptive coding scheme;

Figure 4 shows a schematic illustration of an adaptive coding scheme in the full-rate channel;

Figure 5 shows a schematic illustration of an adaptive coding scheme in the half-rate channel; and

Figure 6 shows an outline circuit diagram of a processor unit.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The structure of the mobile radio system shown in Figure 1 corresponds to that of a known GSM mobile radio system which includes a large number of mobile switching centers MSC which are networked with one another and allow access to a landline network PSTN. Furthermore, these mobile switching centers MSC are each connected to at least one base station controller BSC, which also may be formed by a data processing system. Each base station controller BSC is, in turn, connected to at least one base station BS. A base station BS such as this is a radio appliance which can set up a radio link via a radio interface to radio appliances, which are referred to as mobile stations MS.

The range of the signals from a base station essentially defines a radio cell FZ. The allocation of resources such as frequency bands to radio cells, and thus to the data packets to be transmitted, can be controlled by control devices such as the base station controllers BSC. Base stations BS and a base station controller BSC can be combined to form a base station system BSS.

The base station system BSS is, in this case, also responsible for the radio channel administration, the data rate matching, the monitoring of the radio transmission path, handover procedures, connection control and, possibly, for the allocation and signaling of the voice codecs to be used and, if required, transmits

appropriate signaling information to the mobile stations MS. Such signaling information can also be transmitted via signaling channels.

On the basis of the present description, the present invention can also be used for signaling other information items, such as the type of information (data, speech, pictures, etc.) and/or its coding, switching information, using any desired transmission method, such as DECT, WB-CDMA or multimode transmission methods (GSM/WB-CDMA/TD-CDMA) within a UMTS (Universal Mobile Telephony System).

Figure 2 shows a source Q which produces source signals qs which are compressed by a source coder QE, such as the GSM full-rate voice coding, to form symbol sequences composed of symbols. For parametric source coding methods, the source signals qs (for example, voice) produced by the source Q are subdivided into blocks (for example, time frames) and processed separately. The source coder QE produces quantized parameters (for example, voice coefficients), which are also referred to in the following text as symbols in a symbol sequence, and which reflect the characteristics of the source in that particular block in a specific manner (for example, the voice spectrum, filter parameters). After quantization, these symbols have a specific symbol value.

The symbols in the symbol sequence and the corresponding symbol values are mapped onto a sequence of binary code words, each of which have a number of bit positions, via a binary mapping process (allocation rule), which is frequently written as part of the source coding QE. If, for example, these binary code words are further processed successively as a sequence of binary code words, then this results in a sequence of source-coded bit positions which can be embedded in a frame structure.

Methods which will not be explained here are used to considerably reduce, for example, the original rate of a telephone voice signal (65 kbps μlaw, 104 kbps linear PCM) (approximately 5 kbps - 13 kbps, depending on the coding method). Errors in this bit stream have different effects on the voice quality after decoding. Errors in some bits lead to incomprehension or loud noises, while errors in other

bits are scarcely perceptible. This leads to the bits being subdivided, after the source coder QE, into classes, which generally also have different protection against errors (for example: GSM full-rate codec: Class 1a, 1b and 2). After source coding has been carried out in such a manner, source bits or data bits db are produced, structured in frames, at a source bit rate which depends on the type of source coding.

In mobile radio systems, convolution codes have been found to be efficient codes for subsequent channel coding. If the block length is long, these convolution codes have a high error correction capability and can be decoded with reasonable complexity. For the purposes of example, only rate 1/n convolution codes are dealt with in the following text. A convolution coder with a memory m produces n code bits via a register from the last m+1 data bits.

As already explained above, bits are subdivided into classes during voice coding, and these classes are protected against errors in different ways. This is done by using different rates during convolution coding. Rates greater than ½ are achieved by puncturing.

A new voice and channel coding standard for existing GSM is currently being produced by the Standardization Group for Mobile Radio Systems in Europe (ETSI). In this standard, the voice is intended to be source-coded and different rates using different coding modes, and the channel coding will be adapted appropriately so that the source-coded bit sequences will be coded with respect to channel interference in a channel coder CE, such as a convolution coder, in such a manner that the gross bit rate will still be 22.8 kbps (full-rate mode) or 11.4 kbps (half-rate mode). The particular source bit rate in this case varies depending on the voice (pause, hiss volume, the voice volume itself, strong or weak voice, etc.), as a function of the channel conditions (good, poor channel) and as a function of network conditions (overloading, compatibility, etc.). The channel coding is adapted in a corresponding manner. The particular rate (for example, by virtue of the particular code mode) being used and/or further information are/is transmitted as mode bits mb within the same frame.

As is illustrated in Figure 3, for hierarchical coding purposes, the first portion of the data bits db1 is coded in the same way for all the source bit rates, voice coding rates and code modes being used. This first portion db1 may be the approximately 5·m first source bits. The trellis for this first portion is then set up in the channel decoder QD, and the mode bits are decided on first of all. The particular voice rate or the particular code mode is determined from these mode bits mb, and the second portion of the data bits db2 is also decoded in accordance with the decoding method used for this rate or this code mode.

The first portion or another portion of the data bits db1 can also be channel-coded in a standard manner together with the mode bits mb, independently of the nature of the source coding, that is to say a first portion of the data bits db1 is channel-coded together with the mode bits mb independently of the particular code mode and, to be precise, using a standard method for all the code modes.

This will be explained using a simple example with reference to Figure 3: A source coding method produces frames or blocks using two different code modes with a length of 140 data bits db (case 1) or 100 data bits db (case 2), respectively. A mode bit mb which can also be transmitted in the same frame is intended to indicate which of the two block lengths has just been generated by the source coder QD. After channel coding, a frame with the length of 303 bits is intended to be produced in both cases, which necessarily leads to different channel coding methods, at least with regard to the rate, for the two cases. It is now proposed, in both cases, for a first portion of the data bits db1, for example the first 20 bits, to be channel-coded in a standard manner, for example with regard to the rate (rate 1/3), the convolution codes used, the generator polynomials used or the memory used, and for the matching to the standard frame length of 303 bits to be carried out by using different rates (½ rate for case 1; 1/3 rate for case 2) in the channel coding for the second portion of the 120 (case 1) or 80 (case 2) data bits db2.

In one embodiment of the present invention, the mode bit or bits mb is or are to be channel-coded, in particular convolution-coded in a standard manner together with the first portion of the data bits db1 in both cases; for example, with

regard to the rate (rate 1/3) of the convolution codes used, of the generator polynomials used, or of the memory used.

During decoding, the trellis for a convolution decoder can be set up for the first 21 bits (one mode bit mb + 20 first data bits db1) of a convolution decoder, without knowing what data block length has been used for the coding. If the trellis is set up over this length, the first bit (the mode bit mb) can be determined. The length of influence of the code is taken into account in the process, and the error rate is thus considerably lower than if the trellis were to be set up only for this first mode bit. Once this mode bit has been determined, the block length being used or the code mode used is also known, and the second portion of the data bits db2 is decoded at the 1/2 rate or the 1/3 rate as a function of this.

The decoding complexity is, thus, only insignificantly greater than when decoding only one mode. A systematic convolution code can be used for poor channels, in order to keep the error rate below the channel error rate. A recursive code can be used in order to achieve very good correction characteristics in good channels. For good channels, the error rate is higher than with a non-systematic non-recursive convolution code (previous GSM). However, this is evident only for an error rate of 10^{-4} or less. In this region, any errors which occur can be identified and disguised; the voice quality is not adversely affected.

A scheme for both the half-rate channel and the full-rate channel will be described in the following text.

Figure 4 shows the scheme for the full-rate channel (FR): the voice coding generates four different rates at 13.3 kbps (code mode 1), 9.5 kbp/s (code mode 2), 8.1 kbp/s (code mode 3) and 6.3 kbp/s (code mode 4). The coding is carried out in frames or blocks with a duration of 20 ms. In addition, a CRC with 4 bits is added before the convolution coding in code mode 2, and 2 CRCs each of 3 bits, are added in both code modes 3 and 4. This leads to block lengths of 266 bits db (code mode 1), 199 bits db (code mode 2), 168 bits db (code mode 3) and 132 bits db (code mode 4). Three code mode bits mb are provided at the start of each block or frame in order to signal the particular code mode and in order to transmit further

signaling information. A recursively systematic convolution code at rates of 1/2 and 1/3 is used for the coding. 1/4 and 1/5 rates are produced by bit repetition, and higher rates by puncturing. Once again, the mode bits mb and a first portion of the data bits db1 are channel-coded in the same way for all four code modes, with the mode bits mb always being channel-coded at a rate 1/5, and the first portion of the data bits db1 always being channel-coded at a rate 1/3 or 1/4.

Figure 5 shows the scheme for the half-rate channel (HR): the already explained principle of the same decoding of the mode bits and of the first data bits is also implemented for the half-rate channel. Only the code modes 3 (8.1 kbps) and 4 (6.3 kbps) are used there, with the rate being increased to 11.4 kbps by channel coding. Since fewer code modes are used, 2 mode bits are sufficient in the half-rate channel. The convolution coder used is the same as that in the full-rate codec, but it is not terminated.

As illustrated in Figure 2, bit sequences x or code bits which are channel-coded such as they are matched to the source coding are processed further in a modulator, which is not illustrated, and are then transmitted via a transmission path CH. Interference, such as fading or noise, occurs during transmission.

The transmission path CH is located between a transmitter and a receiver. The receiver may contain an antenna, which is not illustrated, for receiving the signals transmitted via the transmission path CH, a sampling device, a demodulator for demodulating the signals, and an equalizer for elimination of intersymbol interference. Once again, for simplicity reasons, these devices have not been shown in Figure 1. Figure 1 also does not show any possible interleaving and deinterleaving.

The equalizer emits received values from a received sequence y. Owing to interference during transmission on the transmission path CH, the received values have values other than "+1" and "-1".

The channel coding is reversed in a channel decoder CD. The Viterbi algorithm is advantageously used for decoding convolution codes.

Depending on the convolution code memory m, the length of influence during decoding is approximately 5·m. The aim of this is to express the fact that, in general, up to this length of influence, errors in the code can still be corrected. The particular information bit is not corrected by any further removed code bits in the block to be decoded.

The trellis of the decoder is set up approximately as far as the ***5·m*** removed data bit in order to achieve as low an error rate as possible for the first bit in a decoder. A decision on the first bit is then made. In a system with different source coder rates, the source coding of the first ***5·m*** bits is also generally different. As such, the source decoding for these bits is also different, and the decoding process therefore must be carried out differently depending on the source coding rate being used.

Once channel decoding CD has been carried out, this results in the received mode bits <u>m</u>b and data bits <u>db</u> and source decoding QD is carried out to produce received source signals qs, which are emitted at the information sink S.

In embodiment variants of the present invention, other information, particularly control or signaling information, can also be transmitted via the mode bits, such as channel status information or responses to the signaling information (back-channel), information relating to the description of the coding used or the decoding to be used or other information which can be used for decoding the first information items.

Figure 6 shows a processor unit PE which may, in particular, be included in a communications device such as a base station BS or mobile station MS. It includes a control device STE, which essentially includes a programmable microcontroller, and a processing device VE, which includes a processor, particularly a digital signal processor, both of which may have write and read access to memory modules SPE.

The microcontroller controls and monitors all the major elements and functions in a functional unit which includes the processor unit PE. The digital signal processor, a part of the digital signal processor, or a specific processor is

responsible for carrying out the voice coding and voice decoding. The choice of voice codec can also be made by the microcontroller, or by the digital signal processor itself.

An input/output interface I/O is used for inputting/outputting user data or control data, for example, to a control unit MMI, which may include a keyboard and/or a display.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

ABSTRACT OF THE DISCLOSURE

A method and system for channel coding and decoding of information structured in frames, wherein first information items and second information items for describing the coding of first information items are transmitted within a frame, with a first portion of the first information items being channel-coded in standard manner, independently of the nature of the coding, for different types of coding, a first portion of the first information items also being used for decoding second information items.

In the claims:

On page 17, cancel line 1, and substitute the following left-hand justified heading therefor:

We Claims as Our Invention:

Please cancel claims 1-9, without prejudice, and substitute the following claims therefor:

13. A method for channel coding and decoding of data structured in frames, the method comprising the steps of:

selecting a particular code mode from a large number of possible code modes;

source-coding data bits, contained in a frame, in accordance with the particular code mode;

identifying the particular code mode via at least one mode bit contained in the frame; and

channel-coding a first portion of the data bits and the at least one mode bit in a standard manner independently of the particular code mode.

14. A method for channel coding and decoding of data structured in frames as claimed in claim 13, wherein the step of selecting the particular code mode includes matching the particular code mode to at least one of a quality of a transmission channel and a network load.

15. A method for channel coding and decoding of data structured in frames as claimed in claim 13, wherein the at least one mode bit contains at least one of signaling information and information for describing reception quality.

16. A method for channel coding and decoding of data structured in frames as claimed in claim 13, the method further comprising the steps of:

using convolution codes for the step of channel coding; and

selecting the first portion of the data bits as a function of a length of influence of the convolution code.

17. A method for channel coding and decoding of data structured in frames as claimed in claim 13, the method further comprising the step of:

using the first portion of the channel-coded data bits for channel decoding of the at least one mode bit.

18. A method for channel coding and decoding of data structured in frames as claimed in claim 17, the method further comprising the step of:

using knowledge that the first portion of the data bits is channel-coded in a standard manner for different code modes in the process of decoding.

19. A method for channel coding and decoding of data structured in frames as claimed in claim 17, wherein the at least one mode bit is channel-decoded only once.

20. A system for channel coding and decoding of data structured in frames, comprising:

a frame containing data bits which are source-coded in accordance with a particular code mode, the particular code mode being selected from a large number of possible code modes, the frame further containing at least one mode bit for identifying the particular code mode of the data bits; and

a processor unit, via which a first portion of the data bits and the at least one mode bit are channel-coded in a standard manner independently of the particular code mode.

21. A system for channel coding and decoding of data structured in frames as claimed in claim 20, wherein, via the processor unit, the first portion of the channel-coded data bits is also used for channel decoding the at least one mode bit.

REMARKS

The present amendment makes editorial changes and corrects typographical errors in the specification, which includes the Abstract, in order to conform the specification to the requirements of United States Patent Practice. No new matter is added thereby. Attached hereto is a marked-up version of the changes made to the specification by the present amendment. The attached page is captioned "**Version With Markings To Show Changes Made**".

In addition, the present amendment cancels original claims 1-9 in favor of new claims 13-21. Claims 13-21 have been presented solely because the revisions by red-lining and underlining which would have been necessary in claims 1-9 in order to present those claims in accordance with preferred United States Patent Practice would have been too extensive, and thus would have been too burdensome.

The present amendment is intended for clarification purposes only and not for substantial reasons related to patentability pursuant to 35 USC §§103, 102, 103 or 112. Indeed, the cancellation of claims 1-9 does not constitute an intent on the part of the Applicants to surrender any of the subject matter of claims 1-9.

Early consideration on the merits is respectfully requested.

Respectfully submitted,

(Reg. No. 39,056)

William E. Vaughan
Bell, Boyd & Lloyd LLC
P.O. Box 1135
Chicago, Illinois 60690-1135
(312) 807-4292
Attorneys for Applicants

VERSIONS WITH MARKINGS TO SHOW CHANGES MADE

In The Specification:

The Specification of the present application, including the Abstract, has been amended as follows:

SPECIFICATION

TITLE

~~**Method and system for channel coding and decoding of information structured in frames**~~

METHOD AND SYSTEM FOR CHANNEL CODING AND DECODING OF INFORMATION STRUCTURED IN FRAMES

BACKGROUND OF THE INVENTION

~~Description~~

Field of the Invention

The present invention relates, generally, to a method and ~~an arrangement~~ a system for channel coding and decoding of information structured in frames and, ~~in~~ more ~~particular~~ particularly, ~~for the purposes of~~ to such a method and system directed to adaptive multirate coding.

Description of the Prior Art

Source signals and source information such as voice, audio, picture and video signals virtually always contain statistical redundancy, that is to say redundant information. This redundancy can be greatly reduced by source coding, thus allowing efficient transmission and storage of the source signal. This reduction in redundancy gets rid of signal contents which were redundant before transmission and are based on prior knowledge of, for example, statistical parameters in the signal profile. The bit rate of the source-coded information (source bits or data bits) is also referred to as the source bit rate. During the source decoding after transmission, these components are added to the signal once again, so that virtually no loss of quality can be verified objectively.

On the other hand, it is normal for signal transmission to add redundancy deliberately by ~~means of~~ channel coding once again, in order largely to correct for

the influence of channel interference on the transmission. Additional redundant bits thus make it possible for the receiver or decoder to identify errors, and possibly also to correct them. The bit rate of the channel-coded information is also referred to as the gross bit rate.

In order to allow information, in particular voice data, picture data or other user data, to be transmitted as efficiently as possible by ~~means of~~ the limited transmission capacities of a transmission medium, in particular a radio interface, this information to be transmitted is ~~thus~~ compressed by ~~means of~~ source coding before transmission, and is protected against channel errors by ~~means of~~ channel coding. Various methods are, in each case, known for doing this. For example, in the GSM (Global System for Mobile Communication) System, voice can be coded by ~~means of~~ a full rate voice codec a half rate voice codec, or an enhanced full rate voice codec.

For the purposes of this application, the terms voice codec or coding also refer to a method for encoding and/or for corresponding decoding~~,~~. ~~which~~ This method may also cover sources and/or channel coding~~,~~ and can ~~also~~ be applied to data other than voice data.

In the course of the further development of the European mobile radio standard GSM, a new Standard for coded voice transmission is being developed, which will allow the overall data rate and the splitting of the data rate between the source coding and channel coding to be set adaptively depending on the channel state and the network conditions (system load). Instead of the voice codecs described above, which have a fixed source bit rate, new voice codecs are intended to be used for this purpose, whose source bit rate is variable and is matched to changing frame conditions for information transmission. The main aims of such AMR (Adaptive Multirate) voice codecs are to achieve landline network quality for voice transmission in various channel conditions, and to ensure optimum distribution of the channel capacity, taking account of specific network parameters. After carrying out a conventional source coding method, the compressed

information is in structured form, in frames, in which case the source bit rate may differ from frame to frame depending on the code mode being used.

In order to achieve standard gross bit rates, the information contained within a frame is channel-coded in a different manner depending on the source bit rate or code mode, ~~in particular~~ particularly at a different rate, in such a manner that the gross bit rate after channel coding corresponds to the selected channel mode (half rate or full rate). For example, such an AMR voice codec can operate using the half rate (HR) channel in good channel conditions and/or in highly loaded radio cells. When the channel conditions are poor, a dynamic change should be made to the full rate (FR) channel, and vice versa. Within such a channel mode (half rate or full rate), various code modes are available for different voice and channel coding rates, and these are likewise selected to match the channel quality (rate adaptation). In the process, the gross bit rate after channel coding remains constant within one channel mode (22.8 kbps for the full rate channel FR and 11.4 kbps for the half-rate channel HR). This is intended to result in the best voice quality, taking account of the changing channel conditions. Thus, with such adaptive coding, different rates are used for voice coding (variable source bit rate) depending on the channel conditions in a transmission path, on the requirements for specific network parameters, and depending on the voice. Since the gross bit rate after channel coding is intended to remain constant, an appropriately adapted variable number of error protection bits are added during channel coding.

In order to decode such variably coded information after transmission, it is helpful for information about the coding method used at the transmission end, ~~in particular~~ particularly the source bit rate and/or the type of channel coding used at the transmission end, to be known at the receiving end. For this purpose, it is possible for certain bits, so-called mode bits, to be generated at the transmission end~~,~~ which, for example, indicate the rate used for source or channel coding.

It is known for the mode bits to be protected and to be transmitted, using a block code, independently of the source bits (data bits). In consequence, these so-called mode bits can be decoded first ~~of all~~, with the source bits being determined

subsequently, depending on this first decoding result. A disadvantage of this method is that the error frequency is relatively high in the mode bits since, particularly in mobile radio channels which are subject to fading, the correction capability in the decoder is low, owing to the short block length.

Alternatively, it is possible to carry out the decoding in a number of steps. To do this, decoding is initially carried out using a first mode, and a CRC (Cyclic Redundancy Check) is used to determine whether this mode was worthwhile. If this is not the case, decoding is carried out using a further mode, and the result is checked once again. This method is repeated with all the modes until a sensible result is obtained. The disadvantage of this method is the high computation complexity, which leads to both an increased power consumption~~,~~ and ~~to~~ a decoding delay.

It is known from US 5537410 for information to be transmitted in a frame in order to indicate a rate.

The present invention is, thus, based on the problem of specifying a method and ~~an arrangement~~ a system for channel coding and ~~for~~ decoding which allows information about the type of coding to be transmitted in a simple manner and reliably.

~~This problem is solved by the features of the independent patent claims. Developments of the invention can be found in the dependent claims.~~

SUMMARY OF THE INVENTION

~~In order to achieve the object,~~ According to the present invention, therefore, a method is specified in which a first portion of first information items, for example user information items, is channel coded in a standard manner independently of the nature of the coding, and for different types of coding.

This ensures that a first portion of the first information items ~~can~~ also can be used for decoding second information items for describing the coding of first information items, and better quality error correction for the second information items can ~~thus~~ be carried out by the increase in the block length, associated ~~with this~~ therewith, of the convolution codes used for coding the second information

items. This allows multiple decoding, using the try and error principle described above, to be avoided.

Information for describing the coding of first information items may, in this case, contain information for describing the source coding and/or the channel coding and/or other first information items for decoding, such as the type of coding (source and/or channel coding of the first information items) or the coding rate (source and/or channel coding of the first information items).

The present invention can be used advantageously, particularly if the coding of the information can be carried out such that it is adapted to different types.

In one ~~refinement~~ embodiment of the present invention, the rate of channel coding of at least a second portion of the first information items, for example the user information, is matched to the quality of the transmission channel and/or to the network load. It is thus possible to match the channel coding to changing frame conditions in a communications system, and to transmit this adaptation at the transmission end to a receiver in a simple and reliable manner.

In further ~~refinements~~ embodiments, the second information items contain signaling information and/or information for describing the reception quality, in order to influence a transmitter as a function of the reception result at that time. It is thus possible to control the transmission of information based on the principle of a control loop.

For channel coding, it is advantageous to use convolution codes, and to match the length of the first portion of the first information items, which is channel-coded in a standard manner, at least approximately to the length of influence of the convolution code being used.

Furthermore, the problem is solved by a method for decoding of information structured in frames, in which a first portion of the first information items is also used for decoding second information items. This allows the coded transmission of second information items with a sufficiently long block length, and avoids complex multiple decoding based on the ~~try~~ trial and error principle described above.

Decoding carried out in a manner such as this is advantageous, in particular, if one of the methods described above has been used to code the information at the transmission end as part of the information transmission process.

The problem is also solved by ~~arrangements~~ systems for channel coding and decoding of information structured in frames, in which a digital signal processor is in each case set up in such a manner that a first portion of the first information items can be channel-coded in a standard manner, independently of the nature of the coding, for different types of coding, and/or in such a manner that a first portion of the first information items can also be used for decoding second information items. These arrangements are particularly suitable for carrying out the methods according to the present invention, or one of ~~their~~ the alternative embodiments ~~developments~~ explained above.

In this case, digital transmission of information is described, in particular. Nevertheless, the present invention can also be applied to the storage of information, since the writing of information to a storage medium and the reading of information from a storage medium corresponds, in terms of the present invention, to the transmission of information and the reception of information.

DESCRIPTION OF THE DRAWINGS

Figure 1 shows an outline circuit diagram of a mobile radio system;

Figure 2 shows a schematic illustration of major elements of a telecommunications transmission chain;

Figure 3 shows a schematic illustration of an adaptive coding scheme;

Figure 4 shows a schematic illustration of an adaptive coding scheme in the full-rate channel;

Figure 5 shows a schematic illustration of an adaptive coding scheme in the half-rate channel; and

Figure 6 shows an outline circuit diagram of a processor unit.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The structure of the mobile radio system shown in Figure 1 corresponds to that of a known GSM mobile radio system which ~~comprises~~ includes a large

number of mobile switching centers MSC which are networked with one another and allow access to a landline network PSTN. Furthermore, these mobile switching centers MSC are each connected to at least one base station controller BSC, which ~~may~~ also may be formed by a data processing system. Each base station controller BSC is, in turn, connected to at least one base station BS. A base station BS such as this is a radio appliance which can set up a radio link via a radio interface to radio appliances, which are referred to as mobile stations MS.

The range of the signals from a base station essentially defines a radio cell FZ. The allocation of resources such as frequency bands to radio cells, and thus to the data packets to be transmitted, can be controlled by control devices such as the base station controllers BSC. Base stations BS and a base station controller BSC can be combined to form a base station system BSS.

The base station system BSS is, in this case, also responsible for the radio channel administration, the data rate matching, the monitoring of the radio transmission path, handover procedures, connection control and, possibly, for the allocation and signaling of the voice codecs to be used and, if required, transmits appropriate signaling information to the mobile stations MS. Such signaling information can also be transmitted via signaling channels.

On the basis of the present description, the present invention can also be used for signaling other information items, such as the type of information (data, speech, pictures, etc.) and/or its coding, switching information, using any desired transmission method, such as DECT, WB-CDMA or multimode transmission methods (GSM/WB-CDMA/TD-CDMA) within a UMTS (Universal Mobile Telephony System).

Figure 2 shows a source Q which produces source signals qs which are compressed by a source coder QE, such as the GSM full-rate voice coding, to form symbol sequences composed of symbols. For parametric source coding methods, the source signals qs (for example, voice) produced by the source Q are subdivided into blocks (for example, time frames)~~,~~ and ~~these are~~ processed separately. The source coder QE produces quantized parameters (for example, voice coefficients),

which are also referred to in the following text as symbols in a symbol sequence, and which reflect the characteristics of the source in that particular block in a specific manner (for example, the voice spectrum, filter parameters). After quantization, these symbols have a specific symbol value.

The symbols in the symbol sequence and the corresponding symbol values are mapped onto a sequence of binary code words, each of which have a number of bit positions, ~~by means of~~ via a binary mapping process (allocation rule), which is frequently written as part of the source coding QE. If, for example, these binary code words are further processed ~~further~~ successively as a sequence of binary code words, then this results in a sequence of source-coded bit positions~~,~~ which can be embedded in a frame structure.

Methods which will not be explained here are used to considerably reduce, for example, the original rate of a telephone voice signal (65 kbps μlaw, 104 kbps linear PCM) (approximately 5 kbps - 13 kbps, depending on the coding method). Errors in this bit stream have different effects on the voice quality after decoding. Errors in some bits lead to incomprehension or loud noises, while errors in other bits are scarcely perceptible. This leads to the bits being subdivided, after the source coder QE, into classes, which generally also have different protection against errors (for example: GSM full-rate codec: Class 1a, 1b and 2). After source coding has been carried out in such a manner, source bits or data bits db are produced, structured in frames, at a source bit rate which depends on the type of source coding.

In mobile radio systems, convolution codes have been found to be efficient codes for subsequent channel coding. If the block length is long, these convolution codes have a high error correction capability and can be decoded with reasonable complexity. For the purposes of example, only rate 1/n convolution codes are dealt with in the following text. A convolution coder with a memory m produces n code bits via a register from the last m+1 data bits.

As already explained above, bits are subdivided into classes during voice coding, and these classes are protected against errors in different ways. This is done

by ~~means of~~ using different rates during convolution coding. Rates greater than ½ are achieved by puncturing.

A new voice and channel coding standard for existing GSM is currently being produced by the Standardization Group for Mobile Radio Systems in Europe (ETSI). In this standard, the voice is intended to be source-coded and different rates using different coding modes, and the channel coding will be adapted appropriately so that the source-coded bit sequences will be coded with respect to channel interference in a channel coder CE, such as a convolution coder, in such a manner that the gross bit rate will still be 22.8 kbps (full-rate mode) or 11.4 kbps (half-rate mode). The particular source bit rate in this case varies depending on the voice (pause, hiss volume, the voice volume itself, strong or weak voice, etc.), as a function of the channel conditions (good, poor channel) and as a function of network conditions (overloading, compatibility, etc.). The channel coding is adapted in a corresponding manner. The particular rate (for example, by virtue of the particular code mode) being used and/or further information are/is transmitted as mode bits mb within the same frame.

As is illustrated in Figure 3, for hierarchical coding purposes, the first portion of the data bits db1 is coded in the same way for all the source bit rates, voice coding rates and code modes being used. This first portion db1 may be the approximately 5·m first source bits. The trellis for this first portion is then set up in the channel decoder QD, and the mode bits are decided on first of all. The particular voice rate or the particular code mode is determined from these mode bits mb, and the second portion of the data bits db2 is also decoded in accordance with the decoding method used for this rate or this code mode.

The first portion or another portion of the data bits db1 can also be channel-coded in a standard manner together with the mode bits mb, independently of the nature of the source coding, that is to say a first portion of the data bits db1 is channel-coded together with the mode bits mb independently of the particular code mode~~,~~ and, to be precise, ~~likewise~~ using a standard method for all the code modes.

This will be explained using a simple example~~,~~ with reference to Figure 3:

A source coding method produces frames or blocks using two different code modes with a length of 140 data bits db (case 1) or 100 data bits db (case 2), respectively. A mode bit mb which can also be transmitted in the same frame is intended to indicate which of the two block lengths has just been generated by the source coder QD. After channel coding, a frame with the length of 303 bits is intended to be produced in both cases, which necessarily leads to different channel coding methods, at least with regard to the rate, for the two cases. It is now proposed, in both cases, for a first portion of the data bits db1, for example the first 20 bits, to be channel-coded in a standard manner, for example with regard to the rate (rate 1/3), the convolution codes used, the generator polynomials used or the memory used, and for the matching to the standard frame length of 303 bits to be carried out by using different rates (½ rate for case 1; 1/3 rate for case 2) in the channel coding for the second portion of the 120 (case 1) or 80 (case 2) data bits db2.

In one embodiment ~~variant~~ of the present invention, the mode bit or bits mb is or are to be channel-coded, in particular convolution-coded in a standard manner together with the first portion of the data bits db1 in both cases~~,~~; for example, with regard to the rate (rate 1/3) of the convolution codes used, of the generator polynomials used, or of the memory used.

During decoding, the trellis for a convolution decoder can be set up for the first 21 bits (one mode bit mb + 20 first data bits db1) of a convolution decoder, without knowing what data block length has been used for the coding. If the trellis is set up over this length, the first bit (the mode bit mb) can be determined. The length of influence of the code is taken into account in the process, and the error rate is thus considerably lower than if the trellis were to be set up only for this first mode bit. Once this mode bit has been determined, the block length being used or the code mode used is also known, and the second portion of the data bits db2 is decoded at the 1/2 rate or the 1/3 rate as a function of this.

The decoding complexity is, thus, only insignificantly greater than when decoding only one mode. A systematic convolution code can be used for poor channels, in order to keep the error rate below the channel error rate. A recursive

code can be used in order to achieve very good correction characteristics in good channels~~,~~. ~~despite this.~~ For good channels, the error rate is higher than with a non-systematic non-recursive convolution code (previous GSM). However, this is evident only for an error rate of 10^{-4} or less. In this region, any errors which occur can be identified and disguised; the voice quality is not adversely affected.

A scheme for both the half-rate channel and the full-rate channel will be described in the following text.

Figure 4 shows the scheme for the full-rate channel (FR): the voice coding generates four different rates at 13.3 kbps (code mode 1), 9.5 kbp/s (code mode 2), 8.1 kbp/s (code mode 3) and 6.3 kbp/s (code mode 4). The coding is carried out in frames or blocks with a duration of 20 ms. In addition, a CRC with 4 bits is added before the convolution coding in code mode 2, and 2 CRCs each of 3 bits, are added in both code modes 3 and 4. This leads to block lengths of 266 bits db (code mode 1), 199 bits db (code mode 2), 168 bits db (code mode 3) and 132 bits db (code mode 4). Three code mode bits mb are provided at the start of each block or frame in order to signal the particular code mode and in order to transmit further signaling information. A recursively systematic convolution code at rates of 1/2 and 1/3 is used for the coding. 1/4 and 1/5 rates are produced by bit repetition, and higher rates by puncturing. Once again, the mode bits mb and a first portion of the data bits db1 are channel-coded in the same way for all four code modes, with the mode bits mb always being channel-coded at a rate 1/5, and the first portion of the data bits db1 always being channel-coded at a rate 1/3 or 1/4.

Figure 5 shows the scheme for the half-rate channel (HR): the already explained principle of the same decoding of the mode bits and of the first data bits is also implemented for the half-rate channel. Only the code modes 3 (8.1 kbps) and 4 (6.3 kbps) are used there, with the rate being increased to 11.4 kbps by channel coding. Since fewer code modes are used, 2 mode bits are sufficient in the half-rate channel. The convolution coder used is the same as that in the full-rate codec, but it is not terminated.

As illustrated in Figure 2, bit sequences x or code bits which are channel-coded such as they are matched to the source coding are processed further in a modulator, which is not illustrated, and are then transmitted via a transmission path CH. Interference, such as fading or noise, occurs during transmission.

The transmission path CH is located between a transmitter and a receiver. The receiver may contain an antenna, which is not illustrated, for receiving the signals transmitted via the transmission path CH, a sampling device, a demodulator for demodulating the signals, and an equalizer for elimination of intersymbol interference. Once again, for simplicity reasons, these devices have not been shown in Figure 1. ~~The figure~~ Figure 1 also does not show any possible interleaving and deinterleaving.

The equalizer emits received values from a received sequence y. Owing to interference during transmission on the transmission path CH, the received values have values other than "+1" and "-1".

The channel coding is reversed in a channel decoder CD. The Viterbi algorithm is advantageously used for decoding convolution codes.

Depending on the convolution code memory m, the length of influence during decoding is approximately 5·m. The aim of this is to express the fact that, in general, up to this length of influence, errors in the code can still be corrected. The particular information bit is not corrected by any further removed code bits in the block to be decoded.

The trellis of the decoder is set up approximately as far as the $\boldsymbol{5 \cdot m}$ removed data bit in order to achieve as low an error rate as possible for the first bit in a decoder. A decision on the first bit is then made. In a system with different source coder rates, the source coding of the first $\boldsymbol{5 \cdot m}$ bits is also generally different. ~~This means that~~ As such, the source decoding for these bits is also different, and the decoding process ~~must~~ therefore must be carried out differently depending on the source coding rate being used.

Once channel decoding CD has been carried out, this results in the received mode bits mb and data bits db and source decoding QD is carried out to produce received source signals qs, which are emitted at the information sink S.

In embodiment variants of the present invention, other information, ~~in particular~~ particularly control or signaling information, can also be transmitted ~~by means of~~ via the mode bits, such as channel status information or responses to the signaling information (back-channel), information relating to the description of the coding used or the decoding to be used or other information which can be used for decoding the first information items.

Figure 6 shows a processor unit PE which may, in particular, be included in a communications device such as a base station BS or mobile station MS. It includes a control device STE, which essentially ~~comprises~~ includes a programmable microcontroller, and a processing device VE, which ~~comprises~~ includes a processor, ~~in particular~~ particularly a digital signal processor, both of which may have write and read access to memory modules SPE.

The microcontroller controls and monitors all the major elements and functions in a functional unit which includes the processor unit PE. The digital signal processor, a part of the digital signal processor, or a specific processor is responsible for carrying out the voice coding and voice decoding. The choice of voice codec can also be made by the microcontroller, or by the digital signal processor itself.

An input/output interface I/O is used for inputting/outputting user data or control data, for example, to a control unit MMI, which may include a keyboard and/or a display.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

~~Abstract~~

ABSTRACT OF THE DISCLOSURE

~~Method and arrangement for channel coding and decoding of information structured in frames~~

A method and system for channel coding and decoding of information structured in frames, wherein first information items and second information items for describing the coding of first information items are transmitted within a frame, with a first portion of the first information items being channel-coded in standard manner, independently of the nature of the coding, for different types of coding~~.~~, ~~A~~ a first portion of the first information items ~~is~~ also being used for decoding second information items.

~~Figure 3~~

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Description

Method and arrangement for channel coding and decoding of information structured in frames

The invention relates to a method and an arrangement for channel coding and decoding of information structured in frames, in particular for the purposes of adaptive multirate coding.

Source signals and source information such as voice, audio, picture and video signals virtually always contain statistical redundancy, that is to say redundant information. This redundancy can be greatly reduced by source coding, thus allowing efficient transmission and storage of the source signal. This reduction in redundancy gets rid of signal contents which were redundant before transmission and are based on prior knowledge of, for example, statistical parameters in the signal profile. The bit rate of the source-coded information (source bits or data bits) is also referred to as the source bit rate. During the source decoding after transmission, these components are added to the signal once again, so that virtually no loss of quality can be verified objectively.

On the other hand, it is normal for signal transmission to add redundancy deliberately by means of channel coding once again, in order largely to correct for the influence of channel interference on the transmission. Additional redundant bits thus make it possible for the receiver or decoder to identify errors, and possibly also to correct them. The bit rate of the channel-coded information is also referred to as the gross bit rate.

AMENDED SHEET

In order to allow information, in particular voice data, picture data or other user data, to be

transmitted as efficiently as possible by means of the limited transmission capacities of a transmission medium, in particular a radio interface, this information to be transmitted is thus compressed by means of source coding before transmission, and is protected against channel errors by means of channel coding. Various methods are in each case known for doing this. For example, in the GSM (Global System for Mobile Communication) System, voice can be coded by means of a full rate voice codec a half rate voice codec, or an enhanced full rate voice codec.

For the purposes of this application, the terms voice codec or coding also refer to a method for encoding and/or for corresponding decoding, which method may also cover sources and/or channel coding, and can also be applied to data other than voice data.

In the course of the further development of the European mobile radio standard GSM, a new Standard for coded voice transmission is being developed, which will allow the overall data rate and the splitting of the data rate between the source coding and channel coding to be set adaptively depending on the channel state and the network conditions (system load). Instead of the voice codecs described above, which have a fixed source bit rate, new voice codecs are intended to be used for this purpose, whose source bit rate is variable and is matched to changing frame conditions for information transmission. The main aims of such AMR (Adaptive Multirate) voice codecs are to achieve landline network quality for voice transmission in various channel conditions, and to ensure optimum distribution of the channel capacity, taking account of specific network

parameters. After carrying out a conventional source coding method, the compressed information is in structured form, in frames, in which case the source bit rate may differ from frame to frame depending on the code mode being used.

In order to achieve standard gross bit rates, the information contained within a frame is channel-coded in a different manner depending on the source bit rate or code mode, in particular at a different rate, in such a manner that the gross bit rate after channel coding corresponds to the selected channel mode (half rate or full rate). For example, such an AMR voice codec can operate using the half rate (HR) channel in good channel conditions and/or in highly loaded radio cells. When the channel conditions are poor, a dynamic change should be made to the full rate (FR) channel, and vice versa. Within such a channel mode (half rate or full rate), various code modes are available for different voice and channel coding rates, and these are likewise selected to match the channel quality (rate adaptation). In the process, the gross bit rate after channel coding remains constant within one channel mode (22.8 kbps for the full rate channel FR and 11.4 kbps for the half-rate channel HR). This is intended to result in the best voice quality, taking account of the changing channel conditions. Thus, with such adaptive coding, different rates are used for voice coding (variable source bit rate) depending on the channel conditions in a transmission path, on the requirements for specific network parameters, and depending on the voice. Since the gross bit rate after channel coding is intended to remain constant, an appropriately adapted variable number of error protection bits are added during channel coding.

In order to decode such variably coded information after transmission, it is helpful for information about the coding method used at the transmission end, in particular the source bit rate and/or the type of

channel coding used at the transmission end, to be known at the receiving end. For this purpose, it is possible for certain bits, so-called mode bits,

to be generated at the transmission end, which, for example, indicate the rate used for source or channel coding.

It is known for the mode bits to be protected and to be transmitted using a block code, independently of the source bits (data bits). In consequence, these so-called mode bits can be decoded first of all, with the source bits being determined subsequently, depending on this first decoding result. A disadvantage of this method is that the error frequency is relatively high in the mode bits since, particularly in mobile radio channels which are subject to fading, the correction capability in the decoder is low, owing to the short block length.

Alternatively, it is possible to carry out the decoding in a number of steps. To do this, decoding is initially carried out using a first mode, and a CRC (Cyclic Redundancy Check) is used to determine whether this mode was worthwhile. If this is not the case, decoding is carried out using a further mode, and the result is checked once again. This method is repeated with all the modes until a sensible result is obtained. The disadvantage of this method is the high computation complexity, which leads to an increased power consumption, and to a decoding delay.

It is known from US 5537410 for information to be transmitted in a frame in order to indicate a rate.

The invention is thus based on the problem of specifying a method and an arrangement for channel coding and for decoding which allows information about

the type of coding to be transmitted in a simple manner and reliably.

This problem is solved by the features of the independent patent claims. Developments of the invention can be found in the dependent claims.

In order to achieve the object, a method is specified in which a first portion of first information items, for example user information items, is channel coded in a standard manner independently of the nature of the coding, for different types of coding.

This ensures that a first portion of the first information items can also be used for decoding second information items for describing the coding of first information items, and better quality error correction for the second information items can thus be carried out by the increase in the block length, associated with this, of the convolution codes used for coding the second information items. This allows multiple decoding, using the try and error principle described above, to be avoided.

Information for describing the coding of first information items may in this case contain information for describing the source coding and/or the channel coding and/or other first information items for decoding, such as the type of coding (source and/or channel coding of the first information items) or the coding rate (source and/or channel coding of the first information items).

The invention can be used advantageously, particularly if the coding of the information can be carried out such that it is adapted to different types.

In one refinement of the invention, the rate of channel coding of at least a second portion of the first information items, for example the user information, is

matched to the quality of the transmission channel and/or to the network load. It is thus possible to match the channel coding to changing frame conditions in a communications system,

and to transmit this adaptation at the transmission end to a receiver in a simple and reliable manner.

In further refinements, the second information items contain signaling information and/or information for describing the reception quality, in order to influence a transmitter as a function of the reception result at that time. It is thus possible to control the transmission of information based on the principle of a control loop.

For channel coding, it is advantageous to use convolution codes, and to match the length of the first portion of the first information items, which is channel-coded in a standard manner, at least approximately to the length of influence of the convolution code being used.

Furthermore, the problem is solved by a method for decoding of information structured in frames, in which a first portion of the first information items is also used for decoding second information items. This allows the coded transmission of second information items with a sufficiently long block length, and avoids complex multiple decoding based on the try and error principle described above.

Decoding carried out in a manner such as this is advantageous in particular if one of the methods described above has been used to code the information at the transmission end as part of the information transmission process.

The problem is also solved by arrangements for channel

coding and decoding of information structured in frames, in which a digital signal processor is in each case set up in such a manner that a first portion of the first information items can be channel-coded in a standard manner, independently of the nature of the coding, for different types of coding,

and/or in such a manner that a first portion of the first information items can also be used for decoding second information items. These arrangements are particularly suitable for carrying out the methods according to the invention, or one of their developments explained above.

Exemplary embodiments will be described and explained in the following text with reference to the following drawings. In this case, digital transmission of information is described, in particular. Nevertheless, the invention can also be applied to the storage of information, since the writing of information to a storage medium and the reading of information from a storage medium corresponds, in terms of the present invention to the transmission of information and the reception of information.

Figure 1 shows an outline circuit diagram of a mobile radio system;

Figure 2 shows a schematic illustration of major elements of a telecommunications transmission chain;

Figure 3 shows a schematic illustration of an adaptive coding scheme;

Figure 4 shows a schematic illustration of an adaptive coding scheme in the full-rate channel;

Figure 5 shows a schematic illustration of an adaptive coding scheme in the half-rate channel;

Figure 6 shows an outline circuit diagram of a processor unit.

The structure of the mobile radio system shown in Figure 1 corresponds to that of a known GSM mobile

radio system which comprises a large number of mobile switching centers MSC which are networked with one another and allow access to a landline network PSTN. Furthermore, these mobile switching centers MSC are each connected to at least one base station controller BSC, which may also be formed by a data processing system. Each base station controller BSC is in turn

connected to at least one base station BS. A base station BS such as this is a radio appliance which can set up a radio link via a radio interface to radio appliances, which are referred to as mobile stations MS.

The range of the signals from a base station essentially defines a radio cell FZ. The allocation of resources such as frequency bands to radio cells, and thus to the data packets to be transmitted, can be controlled by control devices such as the base station controllers BSC. Base stations BS and a base station controller BSC can be combined to form a base station system BSS.

The base station system BSS is in this case also responsible for the radio channel administration, the data rate matching, the monitoring of the radio transmission path, handover procedures, connection control and, possibly, for the allocation and signaling of the voice codecs to be used and, if required, transmits appropriate signaling information to the mobile stations MS. Such signaling information can also be transmitted via signaling channels.

On the basis of the present description, the invention can also be used for signaling other information items, such as the type of information (data, speech, pictures, etc.) and/or its coding, switching information, using any desired transmission method, such as DECT, WB-CDMA or multimode transmission methods (GSM/WB-CDMA/TD-CDMA) within a UMTS (Universal Mobile Telephony System).

Figure 2 shows a source Q which produces source signals qs which are compressed by a source coder QE, such as the GSM full-rate voice coding, to form symbol

sequences composed of symbols. For parametric source coding methods, the source signals qs (for example voice) produced by the source Q are subdivided into blocks (for example time frames), and these are processed separately. The source coder QE produces quantized parameters (for example voice coefficients), which are also referred to in the following text as symbols in a symbol sequence, and which reflect the characteristics of the source in that particular block in a specific manner (for example the voice spectrum, filter parameters). After quantization, these symbols have a specific symbol value.

The symbols in the symbol sequence and the corresponding symbol values are mapped onto a sequence of binary code words, each of which have a number of bit positions, by means of a binary mapping process (allocation rule), which is frequently written as part of the source coding QE. If, for example, these binary code words are processed further successively as a sequence of binary code words, then this results in a sequence of source-coded bit positions, which can be embedded in a frame structure.

Methods which will not be explained here are used to considerably reduce, for example, the original rate of a telephone voice signal (65 kbps µlaw, 104 kbps linear PCM) (approximately 5 kbps - 13 kbps, depending on the coding method). Errors in this bit stream have different effects on the voice quality after decoding. Errors in some bits lead to incomprehension or loud noises, while errors in other bits are scarcely perceptible. This leads to the bits being subdivided after the source coder QE into classes, which generally

also have different protection against errors (for example: GSM full-rate codec: Class 1a, 1b and

2). After source coding has been carried out in such a manner, source bits or data bits db are produced,

structured in frames, at a source bit rate which depends on the type of source coding.

In mobile radio systems, convolution codes have been found to be efficient codes for subsequent channel coding. If the block length is long, these convolution codes have a high error correction capability and can be decoded with reasonable complexity. For the purposes of example, only rate 1/n convolution codes are dealt with in the following text. A convolution coder with a memory m produces n code bits via a register from the last m+1 data bits.

As already explained above, bits are subdivided into classes during voice coding, and these classes are protected against errors in different ways. This is done by means of different rates during convolution coding. Rates greater than ½ are achieved by puncturing.

A new voice and channel coding standard for existing GSM is currently being produced by the Standardization Group for Mobile Radio Systems in Europe (ETSI). In this standard, the voice is intended to be source-coded and different rates using different coding modes, and the channel coding will be adapted appropriately so that the source-coded bit sequences will be coded with respect to channel interference in a channel coder CE, such as a convolution coder, in such a manner that the gross bit rate will still be 22.8 kbps (full-rate mode) or 11.4 kbps (half-rate mode). The particular source bit rate in this case varies depending on the voice (pause, hiss volume, the voice volume itself, strong or weak voice, etc.), as a function of the channel conditions (good, poor channel) and as a function of

network conditions (overloading, compatibility, etc.). The channel coding is adapted in a corresponding manner. The particular rate (for example by virtue of the particular code mode) being used and/or further information are/is transmitted as mode bits mb within the same frame.

As is illustrated in Figure 3, for hierarchical coding purposes, the first portion of the data bits db1 is coded in the same way for all the source bit rates, voice coding rates and code modes being used. This first portion db1 may be the approximately 5·m first source bits. The trellis for this first portion is then set up in the channel decoder QD, and the mode bits are decided on first of all. The particular voice rate or the particular code mode is determined from these mode bits mb, and the second portion of the data bits db2 is also decoded in accordance with the decoding method used for this rate or this code mode.

The first portion or another portion of the data bits db1 can also be channel-coded in a standard manner together with the mode bits mb, independently of the nature of the source coding, that is to say a first portion of the data bits db1 is channel-coded together with the mode bits mb independently of the particular code mode, and to be precise likewise using a standard method for all the code modes.

This will be explained using a simple example, with reference to Figure 3:
A source coding method produces frames or blocks using two different code modes with a length of 140 data bits db (case 1) or 100 data bits db (case 2), respectively. A mode bit mb which can also be transmitted in the same frame is intended to indicate which of the two block lengths has just been generated by the source coder QD. After channel coding, a frame with the length of 303 bits is intended to be produced in both cases, which necessarily leads to different channel coding methods, at least with regard to the rate, for the two cases. It is now proposed, in both cases, for a first portion

of the data bits db1, for example the first 20 bits, to be channel-coded in a standard manner, for example with regard to the rate (rate 1/3),

the convolution codes used, the generator polynomials used or the memory used, and for the matching to the standard frame length of 303 bits to be carried out by using different rates (½ rate for case 1; 1/3 rate for case 2) in the channel coding for the second portion of the 120 (case 1) or 80 (case 2) data bits db2.

In one embodiment variant of the invention, the mode bit or bits mb is or are to be channel-coded, in particular convolution-coded in a standard manner together with the first portion of the data bits db1 in both cases, for example with regard to the rate (rate 1/3) of the convolution codes used, of the generator polynomials used, or of the memory used.

During decoding, the trellis for a convolution decoder can be set up for the first 21 bits (one mode bit mb + 20 first data bits db1) of a convolution decoder, without knowing what data block length has been used for the coding. If the trellis is set up over this length, the first bit (the mode bit mb) can be determined. The length of influence of the code is taken into account in the process, and the error rate is thus considerably lower than if the trellis were to be set up only for this first mode bit. Once this mode bit has been determined, the block length being used or the code mode used is also known, and the second portion of the data bits db2 is decoded at the 1/2 rate or the 1/3 rate as a function of this.

The decoding complexity is thus only insignificantly greater than when decoding only one mode. A systematic convolution code can be used for poor channels, in order to keep the error rate below the channel error rate. A recursive code can be used in order to achieve

very good correction characteristics in good channels, despite this. For good channels, the error rate is higher than with a non-systematic non-recursive convolution code (previous GSM). However, this is evident only for an error rate of 10^{-4} or less. In this region, any errors which occur can be identified and disguised; the voice quality is not adversely affected.

A scheme for both the half-rate channel and the full-rate channel will be described in the following text.

Figure 4 shows the scheme for the full-rate channel (FR): the voice coding generates four different rates at 13.3 kbps (code mode 1), 9.5 kbp/s (code mode 2), 8.1 kbp/s (code mode 3) and 6.3 kbp/s (code mode 4). The coding is carried out in frames or blocks with a duration of 20 ms. In addition, a CRC with 4 bits is added before the convolution coding in code mode 2, and 2 CRCs each of 3 bits, are added in both code modes 3 and 4. This leads to block lengths of 266 bits db (code mode 1), 199 bits db (code mode 2), 168 bits db (code mode 3) and 132 bits db (code mode 4). Three code mode bits mb are provided at the start of each block or frame in order to signal the particular code mode and in order to transmit further signaling information. A recursively systematic convolution code at rates of 1/2 and 1/3 is used for the coding. 1/4 and 1/5 rates are produced by bit repetition, and higher rates by puncturing. Once again, the mode bits mb and a first portion of the data bits db1 are channel-coded in the same way for all four code modes, with the mode bits mb always being channel-coded at a rate 1/5, and the first portion of the data bits db1 always being channel-coded at a rate 1/3 or 1/4.

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Figure 5 shows the scheme for the half-rate channel (HR): the already explained principle of the same decoding of the mode bits

and of the first data bits is also implemented for the half-rate channel. Only the code modes 3 (8.1 kbps) and 4 (6.3 kbps) are used there, with the rate being increased to 11.4 kbps by channel coding. Since fewer code modes are used, 2 mode bits are sufficient in the half-rate channel. The convolution coder used is the same as that in the full-rate codec, but it is not terminated.

As illustrated in Figure 2, bit sequences x or code bits which are channel-coded such as they are matched to the source coding are processed further in a modulator, which is not illustrated, and are then transmitted via a transmission path CH. Interference, such as fading or noise, occurs during transmission.

The transmission path CH is located between a transmitter and a receiver. The receiver may contain an antenna, which is not illustrated, for receiving the signals transmitted via the transmission path CH, a sampling device, a demodulator for demodulating the signals, and an equalizer for elimination of intersymbol interference. Once again, for simplicity reasons, these devices have not been shown in Figure 1. The figure also does not show any possible interleaving and deinterleaving.

The equalizer emits received values from a received sequence y. Owing to interference during transmission on the transmission path CH, the received values have values other than "+1" and "-1".

The channel coding is reversed in a channel decoder CD. The Viterbi algorithm is advantageously used for decoding convolution codes.

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Depending on the convolution code memory m, the length of influence during decoding is approximately 5·m. The aim of this is to express the fact that, in general, up to this length of influence, errors in the code can still be corrected. The particular information bit is not corrected by any further removed code bits in the block to be decoded.

The trellis of the decoder is set up approximately as far as the ***5·m*** removed data bit in order to achieve as low an error rate as possible for the first bit in a decoder. A decision on the first bit is then made. In a system with different source coder rates, the source coding of the first ***5·m*** bits is also generally different. This means that the source decoding for these bits is also different, and the decoding process must therefore be carried out differently depending on the source coding rate being used.

Once channel decoding CD has been carried out, this results in the received mode bits <u>mb</u> and data bits <u>db</u> and source decoding QD is carried out to produce received source signals qs, which are emitted at the information sink S.

In embodiment variants of the invention, other information, in particular control or signaling information, can also be transmitted by means of the mode bits, such as channel status information or responses to the signaling information (back-channel), information relating to the description of the coding used or the decoding to be used or other information which can be used for decoding the first information items.

Figure 6 shows a processor unit PE which may, in particular, be included in a communications device such as a base station BS or mobile station MS. It includes a

control device STE, which essentially comprises a programmable microcontroller, and a processing device VE, which comprises a processor, in particular a digital signal processor, both of which may have write and read access to memory modules SPE.

The microcontroller controls and monitors all the major elements and functions in a functional unit which includes the processor unit PE. The digital signal processor, a part of the digital signal processor, or a specific processor is responsible for carrying out the voice coding and voice decoding. The choice of voice codec can also be made by the microcontroller, or by the digital signal processor itself.

An input/output interface I/O is used for inputting/outputting user data or control data, for example, to a control unit MMI, which may include a keyboard and/or a display.

Patent Claims

1. A method for channel coding of data structured in frames,
   - in which a frame contains data bits (db) which are source-coded in accordance with a code mode, with the code mode being selected from a large number of possible code modes,
   - in which the frame contains at least one mode bit (mb) for identifying the particular code mode of the data bits, and
   - in which a first portion of the data bits (db1) and at least the mode bit (mb) are channel-coded in a standard manner independently of the particular code mode.

2. The method as claimed in claim 1, in which the selection of the particular code mode is matched to the quality of the transmission channel and/or to the network load.

3. The method as claimed in one of the preceding claims, in which the mode bits (mb) contain signaling information and/or information for describing the reception quality.

4. The method as claimed in one of the preceding claims, in which convolution codes are used for channel coding, and the first portion of the data bits, which is channel-coded in a standard manner, is selected as a function of the length of influence of the convolution code.

5. A method for decoding data which is structured in frames and has been coded using a method according to one of claims 1 to 4,
- in which a first portion of the channel-coded data bits (db1) is also used for channel decoding of at least one mode bit (mb).

6. The method as claimed in claim 5, in which knowledge that a first portion of the data bits (db) is channel-coded in a standard manner for different code modes is used.

7. The method as claimed in one of claims 5 to 6, in which the or each mode bit (mb) are channel-decoded only once.

8. An arrangement for channel coding of data structured in frames, with a frame containing data bits (db) which are source-coded in accordance with a code mode, with the code mode being selected from a large number of possible code modes, and with the frame containing at least one mode bit (mb) for identifying the particular code mode of the data bits,
- having a processor unit which is set up in such a manner that a first portion of the data bits (db1) and at least one mode bit (mb) are channel-coded in a standard manner, independently of the particular code mode.

9. An arrangement for decoding data which are structured in frames and have been coded in accordance with a method as claimed in one of claims 1 to 4,
- having a processor unit which is set up in such a manner that a first portion of the channel-coded data bits (db1) is also used for channel decoding at least one mode bit (mb).

Abstract

Method and arrangement for channel coding and decoding of information structured in frames

First information items and second information items for describing the coding of first information items are transmitted within a frame, with a first portion of the first information items being channel-coded in a standard manner, independently of the nature of the coding, for different types of coding. A first portion of the first information items is also used for decoding second information items.

Figure 3

FIG 1

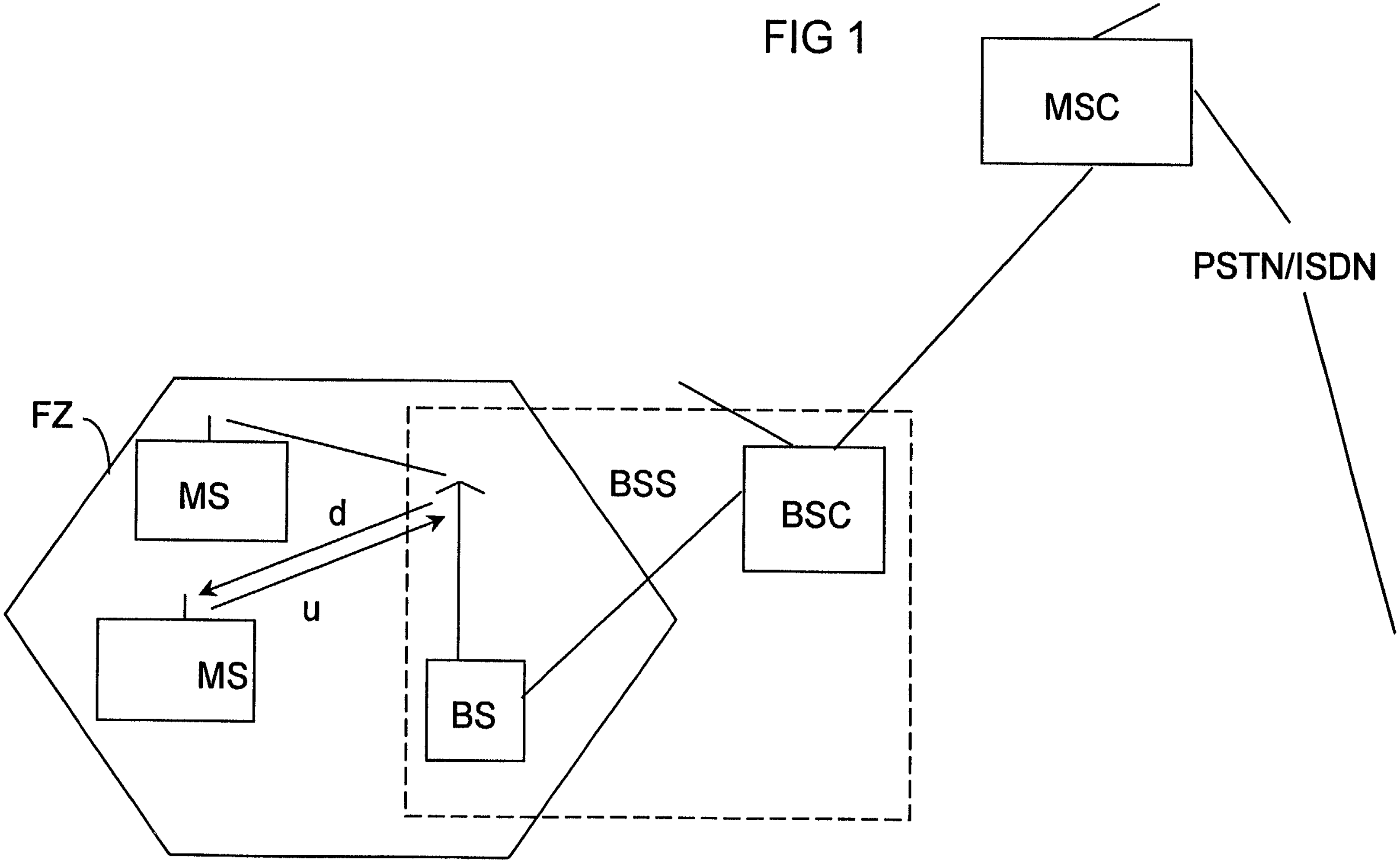

FIG 2

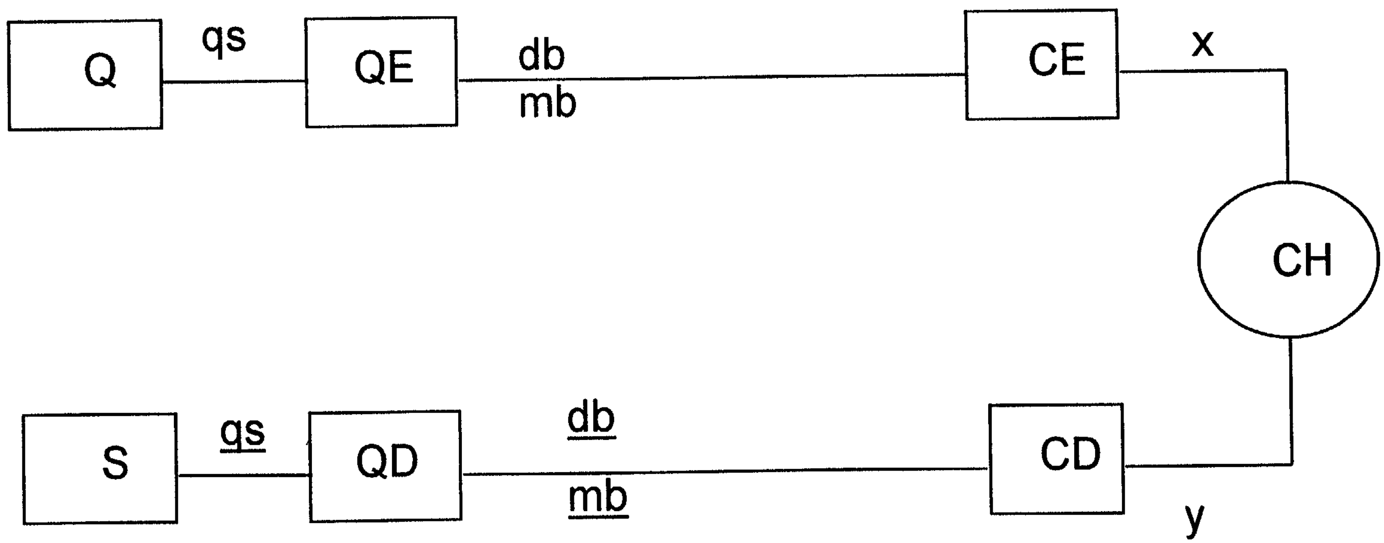

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FIG 3

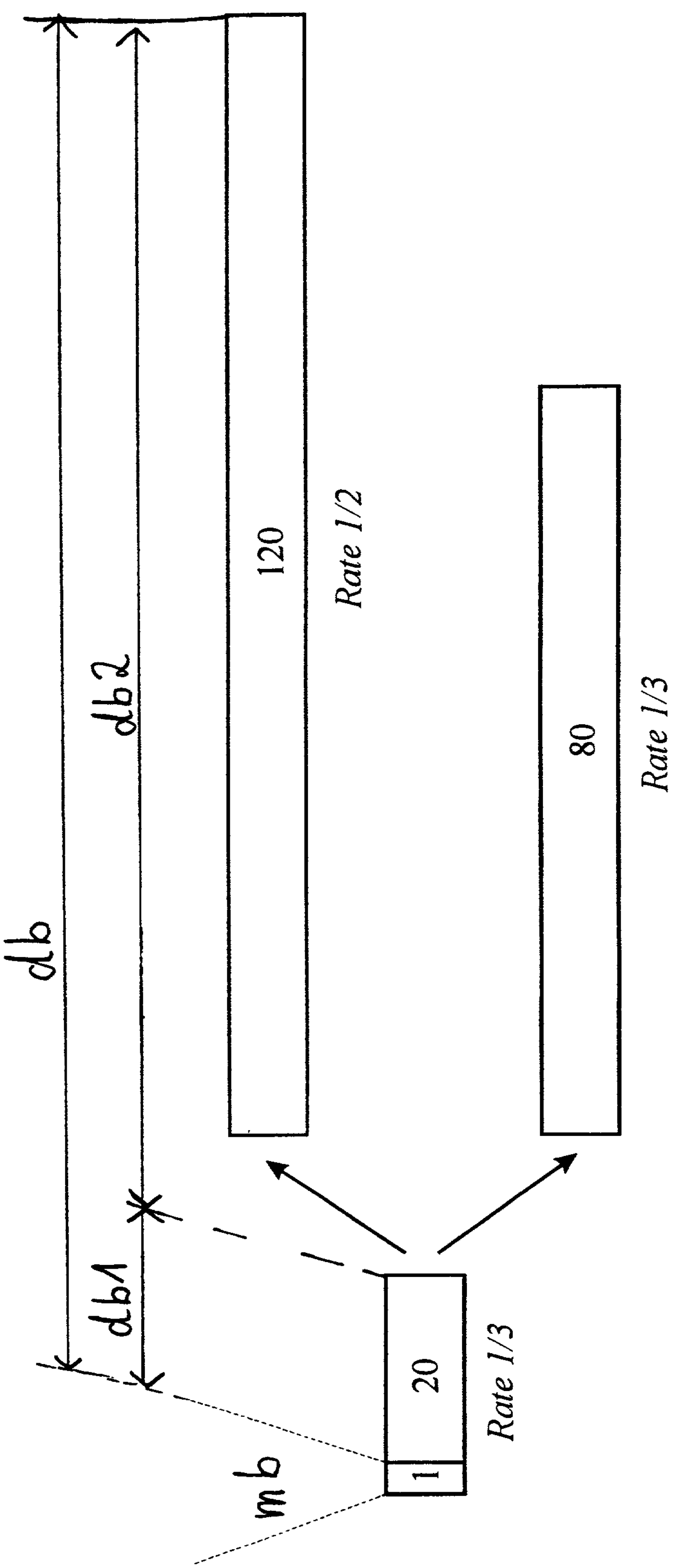
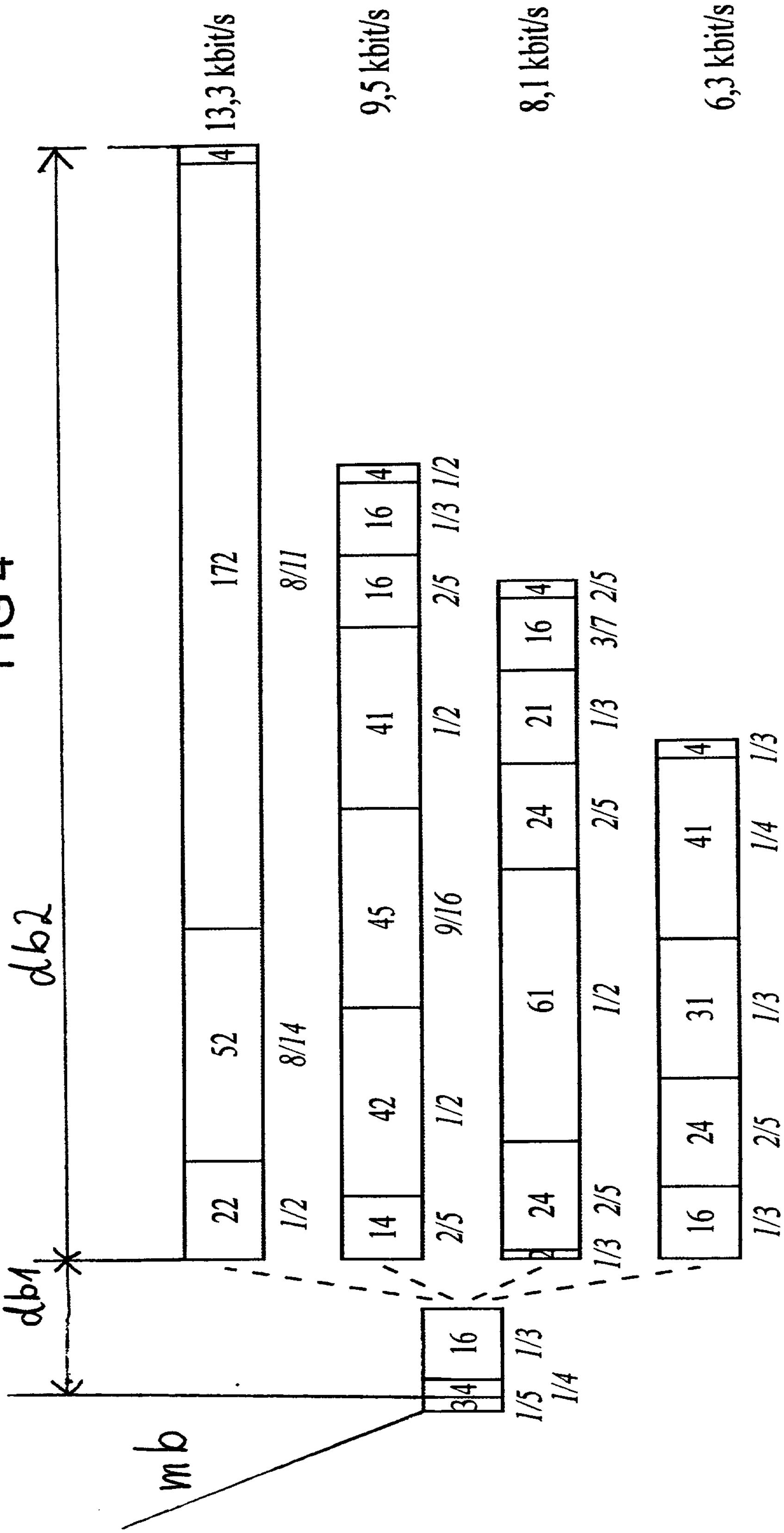


FIG 4


13,3 kbit/s
9,5 kbit/s
8,1 kbit/s
6,3 kbit/s
db2
db1
mb
22
52
172
4
1/2
8/14
8/11
14
42
45
41
16
16
4
2/5
1/2
9/16
1/2
2/5
1/3
1/2
24
61
24
21
16
4
1/3
2/5
1/2
2/5
1/3
3/7
2/5
16
24
31
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1/5
1/4
1/3

FIG 5

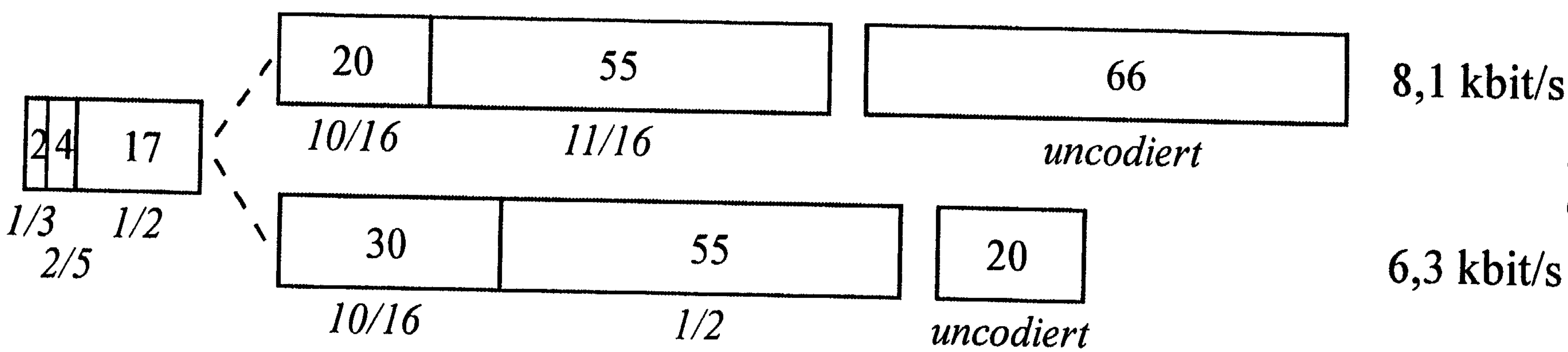

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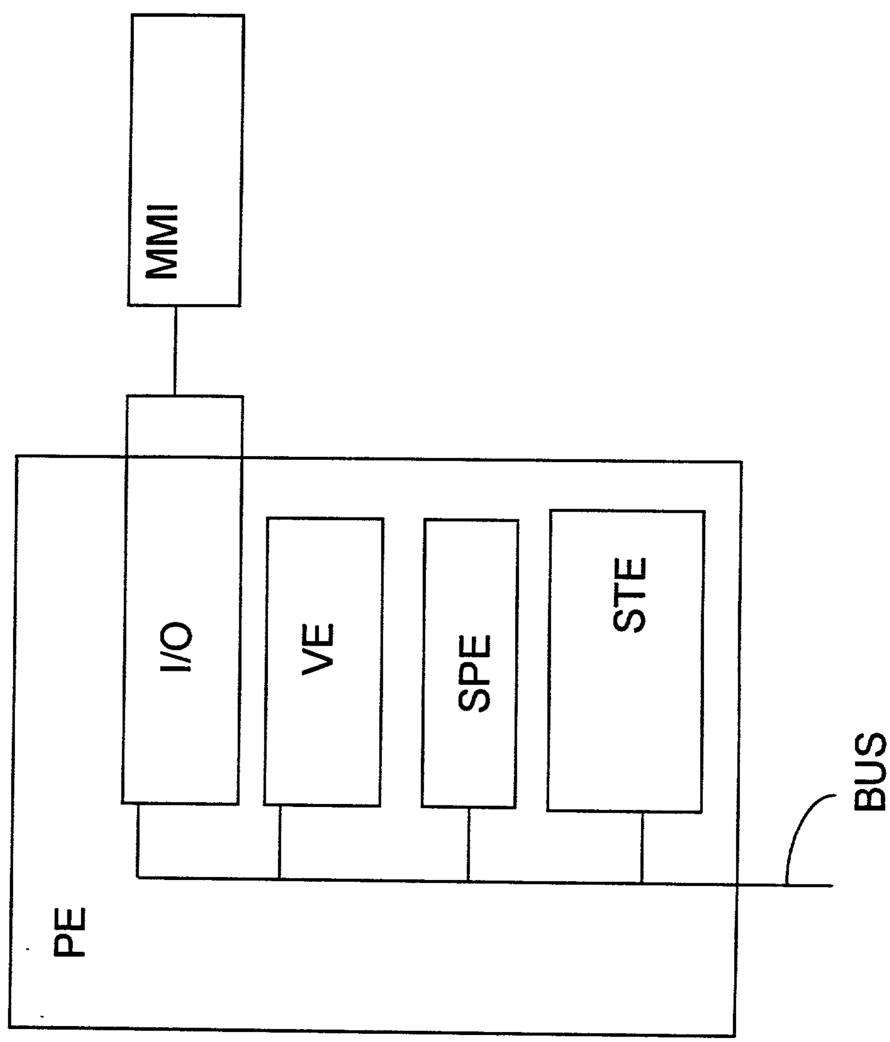
FIG 6



Combined Declaration For Patent Application and Power of Attorney (Continued) (Includes Reference to PCT International Applications) PCT/DE99/03838

ATTORNEY'S DOCKET NO. 112740-218

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) or PCT international application(s) designating the United States of America that is/are listed below and, insofar as the subject mater of each of the claims of this application is not disclosed in that/those prior application(s) in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application(s) and the national or PCT international filing date of this application:

PRIOR U.S. APPLICATIONS OR PCT INTERNATIONAL APPLICATIONS DESIGNATING THE U.S. FOR BENEFIT UNDER 35 U.S.C. 120:

U.S. APPLICATIONS

U.S. APPLICATION NUMBER	U.S. FILING DATE	STATUS (Check one) PATENTED	PENDING	ABANDONED

PCT APPLICATIONS DESIGNATING THE U.S.

PCT APPLICATION NO	PCT FILING DATE	U.S. SERIAL NUMBERS ASSIGNED (if any)	PATENTED	PENDING	ABANDONED

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) Holby M. Abem (P47,372), Robert M. Barrett (30,142), Alan L. Barry (30,819), Thomas C. Basso (46,541), Jeffrey H. Canfield (38,404), Robert W. Connors (46,639), Amy J. Gast (41,773), Timothy L. Harney (38,174), Patricia A. Kane (46,446), Michael S. Leonard (37,557), Edward A. Lehman (22,312), Adam H. Masia (35,602), Dante J. Picciano (33,543), Renato L. Smith (45,117), Maurice E. Teixeira (45,646), William E. Vaughan (39,056), Austin Victor (47,154), and all members of the firm of Bell, Boyd & Lloyd LLC.

Send Correspondence to:

William E. Vaughan
BELL, BOYD & LLOYD LLC
P.O. Box 1135
Chicago, Illinois 60690

Direct Telephone Calls to: 312/807-4292

201	FULL NAME OF INVENTOR	FAMILY NAME HAGENAUER	FIRST GIVEN NAME JOACHIM	SECOND GIVEN NAME	
	RESIDENCE & CITIZENSHIP	CITY 82229 Seefeld	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Germany	
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Peter Rosegger Str. 41	CITY 82229 Seefeld	STATE & ZIP CODE/COUNTRY Germany	
200 202	FULL NAME OF INVENTOR	FAMILY NAME XU	FIRST GIVEN NAME WEN	SECOND GIVEN NAME	
	RESIDENCE & CITIZENSHIP	CITY 82008 Unterhaching DEX	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Germany	
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Bischofschofener Str. 11	CITY 82008 Unterhaching	STATE & ZIP CODE/COUNTRY Germany	
300 203	FULL NAME OF INVENTOR	FAMILY NAME HINDELANG	FIRST GIVEN NAME THOMAS	SECOND GIVEN NAME	
	RESIDENCE & CITIZENSHIP	CITY 81373 Muenchen DEX	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Gemany	
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Kruener Str. 17	CITY 81373 Muenchen	STATE & ZIP CODE/COUNTRY Germany	

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

SIGNATURE OF INVENTOR 201	SIGNATURE OF INVENTOR 202	SIGNATURE OF INVENTOR 203
	[signature]	[signature]
DATE	DATE 2001-09-12	DATE 2001-09-12

COMBINED DECLARATION FOR PATENT APPLICATION AND POWER OF ATTORNEY (Includes Reference to PCT International Applications) PCT/DE99/03838	ATTORNEY'S DOCKET NUMBER 112740-218

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name, I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

METHOD AND SYSTEM FOR CHANNEL CODING AND DECODING OF INFORMATION STRUCTURED IN FRAMES

the specification of which (check only one item below):

☐ is attached hereto.

■ was filed as United States application
Serial No. 09/868,398

on June 18, 2001,

and was amended

on __________(if applicable).

☐ was filed as PCT international application

Number __________

on __________.

and was amended under PCT Article 19

on __________(if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate or of any PCT international application(s) designating at least one country other than the United States of America listed below and have also identified below any foreign application(s) for patent or inventor's certificate or any PCT international application(s) designating at least one country other than the United States of America filed by me on the same subject matter having a filing date before that of the application(s) of which priority is claimed:

PRIOR FOREIGN/PCT APPLICATION(S) AND ANY PRIORITY CLAIMS UNDER 35 U.S.C. 119:

COUNTRY (If PCT indicate "PCT")	APPLICATION NUMBER	DATE OF FILING (day, month, year)	PRIORITY CLAIMED UNDER 35 USC 119
Germany	198 58 393.1	17 December 1998	■ YES ☐ NO
			☐ YES ☐ NO
			☐ YES ☐ NO
			☐ YES ☐ NO
			☐ YES ☐ NO

Combined Declaration For Patent Application and Power of Attorney (Continued) (Includes Reference to PCT International Applications) PCT/DE99/03838

ATTORNEY'S DOCKET NO. 112740-218

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) or PCT International application(s) designating the United States of America that is/are listed below and, insofar as the subject mater of each of the claims of this application is not disclosed in that/those prior application(s) in the manner provided by the first paragraph of Title 35, Untied States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application(s) and the national or PCT international filing date of this application:

PRIOR U.S. APPLICATIONS OR PCT INTERNATIONAL APPLICATIONS DESIGNATING THE U.S. FOR BENEFIT UNDER 35 U.S.C. 120:

U.S. APPLICATIONS

U.S. APPLICATION NUMBER	U.S. FILING DATE	STATUS (Check one): PATENTED	PENDING	ABANDONED

PCT APPLICATIONS DESIGNATING THE U.S.

PCT APPLICATION NO	PCT FILING DATE	U.S. SERIAL NUMBERS ASSIGNED (if any)	PATENTED	PENDING	ABANDONED

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) Holby M. Abern (P47,372), Robert M. Barrett (30,142), Alan L. Barry (30,819), Thomas C. Basso (46,541), Jeffrey H. Canfield (38,404), Robert W. Connors (46,639), Amy J. Gast (41,773), Timothy L. Harney (38,174), Patricia A. Kane (46,446), Michael S. Leonard (37,557), Edward A. Lehman (22,312), Adam H. Masia (35,602), Dante J. Picciano (33,543), Renato L. Smith (45,117), Maurice E. Teixeira (45,646), William E. Vaughan (39,056), Austin Victor (47,154), and all members of the firm of Bell, Boyd & Lloyd LLC.

Send Correspondence to:	Direct Telephone Calls to:
William E. Vaughan BELL, BOYD & LLOYD LLC P.O. Box 1135 Chicago, Illinois 60690	312/807-4292

1 PD 201	FULL NAME OF INVENTOR	FAMILY NAME HAGENAUER	FIRST GIVEN NAME JOACHIM	SECOND GIVEN NAME
	RESIDENCE & CITIZENSHIP	CITY 82229 Seefeld DEX	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Germany
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Peter Rosegger Str. 41	CITY 82229 Seefeld	STATE & ZIP CODE/COUNTRY Germany
202	FULL NAME OF INVENTOR	FAMILY NAME XU	FIRST GIVEN NAME WEN	SECOND GIVEN NAME
	RESIDENCE & CITIZENSHIP	CITY 82008 Unterhaching	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Germany
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Bischofschofener Str. 11	CITY 82008 Unterhaching	STATE & ZIP CODE/COUNTRY Germany
203	FULL NAME OF INVENTOR	FAMILY NAME HINDELANG	FIRST GIVEN NAME THOMAS	SECOND GIVEN NAME
	RESIDENCE & CITIZENSHIP	CITY 81373 Muenchen	STATE OR FOREIGN COUNTRY Germany	COUNTRY OF CITIZENSHIP Gemany
	POST OFFICE ADDRESS	POST OFFICE ADDRESS Kruener Str. 17	CITY 81373 Muenchen	STATE & ZIP CODE/COUNTRY Germany

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

SIGNATURE OF INVENTOR 201	SIGNATURE OF INVENTOR 202	SIGNATURE OF INVENTOR 203
J. Ha[illegible]		
DATE 25/9/01	DATE	DATE